

# Mobility Management for VoIP on Heterogeneous Networks: Evaluation of Adaptive Schemes

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# OBJECTIVE OF THE PAPER

- Third Generation Partnership Project (3GPP)
  - Seamless mobility among 3G n 2G,WLAN
  - IP Multimedia Subsystems (IMS) - IP-based real-time services
  - Mobile Systems to provide services across different access technologies (differ w.r.t bit rate, delay, packet loss)
- End-to-end adaptive mechanism
  - Support QoS on heterogeneous networks
- Implementation aspect
  - Implemented across the different layers of protocol stack

# PREVIOUS WORK

- “Seamless Internetworking of WLANs and Cellular Networks :Architecture and Performance”
  - Cross-layer mobility support at Link layer
- “Adaptive Streaming on Heterogeneous Networks”
  - Good performance with little overhead for mobile streaming applications
  - Content adaptation at session Layer
- Cross-layer approach to VoIP applications
  - Experimental results summary
  - Performance at network, session, and application layers in a 3G/WLAN scenario



# SCENARIO OVERVIEW

- Vehicular mobility
  - Use high speed wireless connections when available
  - Switch to slower, cellular networks in areas that are outside of a WLAN's coverage
- Smaller areas with less coverage
  - Distribution of access points cannot achieve complete coverage
  - Seamless migration of a real-time session to 3G radio access
- IP Multimedia Subsystems
  - Internet protocols, like SIP as the session control protocol
  - RTP/RTCP to transport real-time media, such as video and audio
  - IMS design - based on IPv6
  - Drawback -Internetworking issues

# SCENARIO OVERVIEW

- Interconnection levels
  - common billing and customer care
  - 3G system-based access control and charging
  - access to 3G system IMS-based services
  - service continuity
  - seamless services
- Requirements met by this approach
  - adequate performance
  - no change to existing network infrastructures
  - no modification to existing protocols



# REQUIREMENTS OF VoIP SERVICES

- No End-to-Mouth(E2M) Delay
  - Less than 150-200 ms
  - 3G cellular networks have values of up to 150-250 ms
- Synchronizing speech playout
  - Network delay -Loss of synchronization
- Steady bit rate
  - Lower bit rate-VoIP application experience congestion - packet loss and service degradation ex:WLAN,UMTS
  - Detect and Adapt
  - HSDPA/HSUPA access networks
- No Ping Pong Effect
  - Effect caused by doing handoff back and forth.

# REQUIREMENTS OF VoIP SERVICES

- Integrated Cross-layer Framework
  - Delay and synchronization issues-Application Layer
  - Bandwidth Requirements -Session Layer
  - Vertical Handoff optimization-Network Layer

# MOBILITY MANAGEMENT FOR VOIP SERVICES

- Mobility in wireless networks - managed at different layers
  - Link Layer Mobility
  - Network Layer Mobility
  - Transport layer Mobility
  - Application Layer Mobility
- Link Layer Mobility
  - Specific to wireless network
  - Ex: WLAN vs UMTS



# MOBILITY MANAGEMENT FOR VOIP SERVICES

- Transport Layer Mobility
  - Achieved by extensions to Transport Protocols
  - Drawbacks
- Application Layer Mobility
  - Application-specific
  - Changes to SIP
  - Drawbacks
- Network Layer Mobility
  - Achieved by extensions to IP Protocol
  - Advantages
    - Better Performance w.r.t solutions implemented at upper layers
    - Transparent-Mobility management

# TERMINOLOGIES

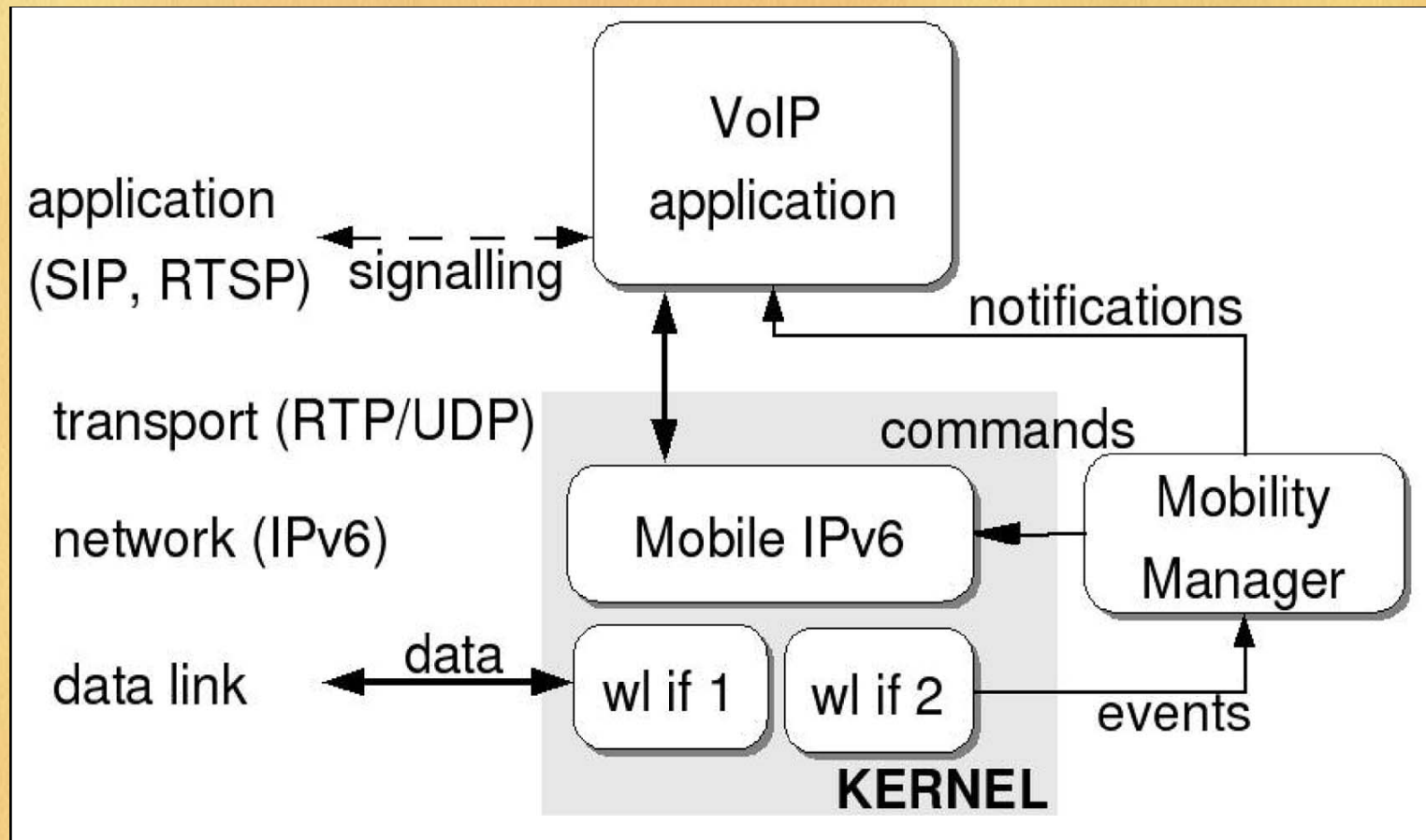
- MN moves through networks-HA tracks current binding of MN
- Binding Update
  - When a handoff takes place, the mobile node sends messages (HA,CN), known as Binding Updates (BU)
- Horizontal Handoff
  - Migration is between homogeneous network
  - when AR becomes unavailable due MH movement
- Vertical Handoff
  - Mobility through heterogeneous networks

# TERMINOLOGIES

- Make before break handoff
  - Facilitates loss-less handoffs- performs all the configuration and signaling steps on the new network before actually leaving the old one
- Break before make handoff
  - Connection disruption
  - Active access router becomes unreachable before handoff execution



# Cross-layer mobility support



# MOBILITY MANAGER

- Efficient vertical handoffs
  - Movement detection
  - Capability of detecting current access network is going to become unusable
- IPv6 method
  - Performance of Mobile IPv6 w.r.t horizontal handoffs -reasonably good
  - To discover router reachability, based on the reception of Router advertisements messages-is not well suited
- Software module(MM) - monitors the link status can react more quickly

# MOBILITY MANAGER

- Mobility Manager (MM) is a user-level application
  - Monitoring connection parameters through appropriate system calls
  - Performing handoff decisions with appropriate timings using MobileIPv6 primitive operations
  - Dispatching notifications about connectivity changes and network parameters
- Scope of MM - Limited to the network layer
  - Application adaptation - performed by adaptive application
- Applications supported by MM
  - Streaming applications and even legacy applications,
  - MM delivers event notifications to an (AMI)-Adapt at session layer



# API

Function name	Input	Output
handoff_notification	pHandoff_f	exit_code
prehandoff_notification	pHandoff_f	exit_code
warning_notification	pWarning_f	exit_code
get_interfaces	void	If_info **info_l
get_curr_interface	void	If_info *info
void (*pHandoff_f) void (*pWarning_f)	If_info *Old>If, If_info *New>If	void

# EXPERIMENTAL RESULTS

- Aim of the experiment
  - Handoff performance at the network layer
- Mechanisms
  - MM Module
  - Basic network layer triggering mechanism of Mobile IPv6 (“L3 triggering”)
- Two types of handoffs are performed
  - Up handoffs (from a faster, but more limited in scope, network to a slower one)
  - Down handoffs
- Handoff delay
  - measured as the time interval between the arrival of the 1<sup>st</sup> packet from the new network and the arrival of the last packet from the old network.

# EXPERIMENTAL RESULTS

	LAN-WLAN (MN)		WLAN-UMTS (MN)		WLAN-UMTS (CN)	
	Delay (ms)	Loss (#)	Delay (ms)	Loss (#)	Delay (ms)	Loss (#)
L3 trigger, 25 packets/s	402	10.0	351	5.0	558	3.0
L3 trigger, 50 packets/s	332	16.6	426	14.3	340	13.7
MM, 25 packets/s	96	1.3	153	0.0	262	2.0
MM, 50 packets/s	89	3.6	167	0.0	234	4.7

Fig :Performance for the Up Handoff among Different Networks

- Difference of packet trip time along the two networks- the packet loss not proportional to the handoff delay
- Packet loss and handoff delay are proportional only for break before make handoffs between networks with comparable packet trip times



# EXPERIMENTAL RESULTS

	Average packet loss (%)
L3 triggering, 25 packets/s	$13.95 \pm 9.40\%$
L3 triggering, 50 packets/s	$13.56 \pm 4.96\%$
MM, 25 packets/s	$4.52 \pm 1.31\%$
MM, 50 packets/s	$2.38 \pm 1.43\%$

Average Packet Loss for 802.11 —  
UMTS Mobility (Four Handoffs)

- Experiments with VoIP conversations of about 180s with four handoffs
- Better performance with the MM stems from several factors:
  - Up handoffs are faster
  - Handoff is executed before the connectivity is lost
  - Ping-pong effects can be avoided

# ADAPTATION AT SESSION LAYER

- VoIP application has two parts
- First part manages call, using a signaling protocol(SIP) for session establishment
  - Signaling protocol determine the session content (codec, audio/video flows, etc.)
  - Session description block
  - Deciding on format
- Second part controls incoming and outgoing media flows

Note: Sessions refers to VoIP sessions

# ADAPTATION AT SESSION LAYER

- Content Adaptation
  - Content – Audio flow
  - Access Network Handoff
  - Actual Bit rate > Nominal Bit rate
  - Mobility-aware VoIP application adapts Session content
    - » By changing the session and
    - » By changing the audio codec
- Audio codecs used for IP telephony
  - G.711 Alaw/Ulaw (64 Kbps)
  - GSM (13 Kbps)
  - G.729 (8 Kbps)
- Session Adaptation
  - Signalling Protocol(SIP)
  - MM agent issues a new SIP INVITE message to the other end of the conversation, specifying new session parameters



# ADAPTATION AT SESSION LAYER

Parameters	
<b>f</b>	Number of samples per second (typical value $f=8000$ )
<b>n</b>	Number of samples contained in each IP packet (n samples make an audio frame)
<b>Lambda</b>	Number of network packets per second ( $=f/n$ )
<b>b0</b>	Bit rate of the codec
<b>h</b>	Total size of the headers (in bytes)
<b>s</b>	packet size (in bytes) $s = h + b_0 * n / 8f$
<b>b</b>	required bit rate (in bps) $b = b_0 + 8fh/n$

- Impact of Packet Overhead
  - Sample application using GSM codec
  - $f = 8000$ ,  $n = 160$  (meaning 50 packets per second) Bit Rate= 42.6 Kbps (nominal=13 Kbps) Payload= 33 bytes
  - $f=8000, n=480$  (each packet contains 480 samples), Bit rate =23 Kbps
- Drawback of increasing sample size
  - Helps reduce uplink bit rate
- Header compression

# ADAPTATION AT SESSION LAYER

- Session content adaptation is performed by the MN
  - Application registers with MM
    - *Whether or how* to do session adaptation
  - Handoff towards n/w with lower bit rate
    - INVITE SIP message
  - Handoff towards n/w with higher bit rate
    - Application waits for the handoff\_notification
  - prehandoff\_notification
    - to trigger the session update before the handoff is actually performed

# ADAPTATION AT SESSION LAYER

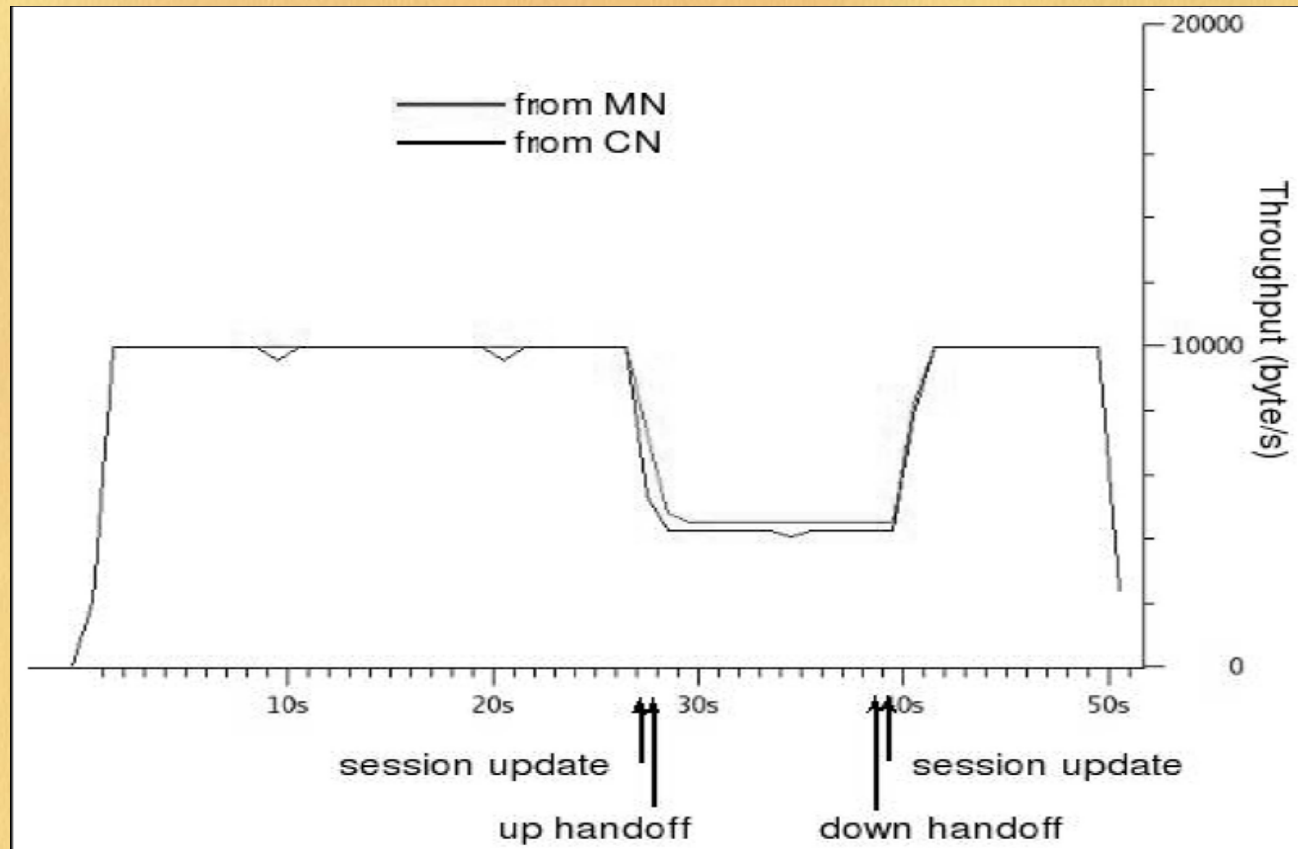


Fig. Shows resulting bit rate in a real WLAN-UMTS-WLAN VoIP session with session content adaptation



# ADAPTATION AT SESSION LAYER-

## Experimental Results

- Initial Values/Settings:
  - VoIP session uses a G.711 Alaw codec(  $b_0 = 64$  Kbps)
  - $f = 8000$ ,  $n = 200$ ,  $\lambda = 40$ ,  $h = 76$ ,  $s = 320$
  - Required BR  $b = 79,200$ bps
- MN sends SIP message
  - To switch to GSM codec  $b_0 = 13$  kbps
  - Required BR ( $b$ ) = 33,000 bps
- Content adaptation is an option to trade off conversation quality against usage of network resources.
- Uses
  - Maintain active connection
  - Used to implement users' policies for allocating dynamically network resources between applications

# ADAPTATION AT THE APPLICATION LAYER

- Jitter Effect and Packet Trip Time
- Working of VoIP
  - Remote peer sends stream of packets
  - Stream received after average network delay ( $d$ )
  - Packet arriving after Playout Time is dropped
  - Receiver side introduces a jitter buffer/playout buffer between the arrival of a packet and its playout, which causes a playout delay
- Trade-off between Playout delay and late packet loss



# ADAPTATION AT THE APPLICATION LAYER

- Playout Algorithm
  - Help keep buffering delay as short as possible while minimizing the number of packets dropped as a consequence of late arrival
- Adaptive Playout Algorithms
  - Vary playout delay during call's lifetime
- Playout algorithms-Categories
  - Reactive algorithms- continuously estimate network delay and jitter
  - Histogram algorithms- maintain a histogram of packet delays
  - Algorithms that monitor the packet-loss ratio or buffer occupancy
- Network delay and jitter
  - Used to proactively change the size of the jitter buffer and playout delay
    - Optimize the loss/delay performance of the application



# ADAPTATION AT THE APPLICATION LAYER-

## Delay across handoffs

- Roaming across heterogeneous networks
  - Variation in delay and jitter
- Jitter buffer introduces a delay in the voice playout
  - $db = S/\text{Lambda}$ ,
  - Total delay (between packet sending and packet playout) becomes not less than  $D = d + db$
- Handoffs to networks with larger( $d$ )
  - Increased delay causes massive packet discarding

# ADAPTATION AT THE APPLICATION LAYER-

## Delay across handoffs

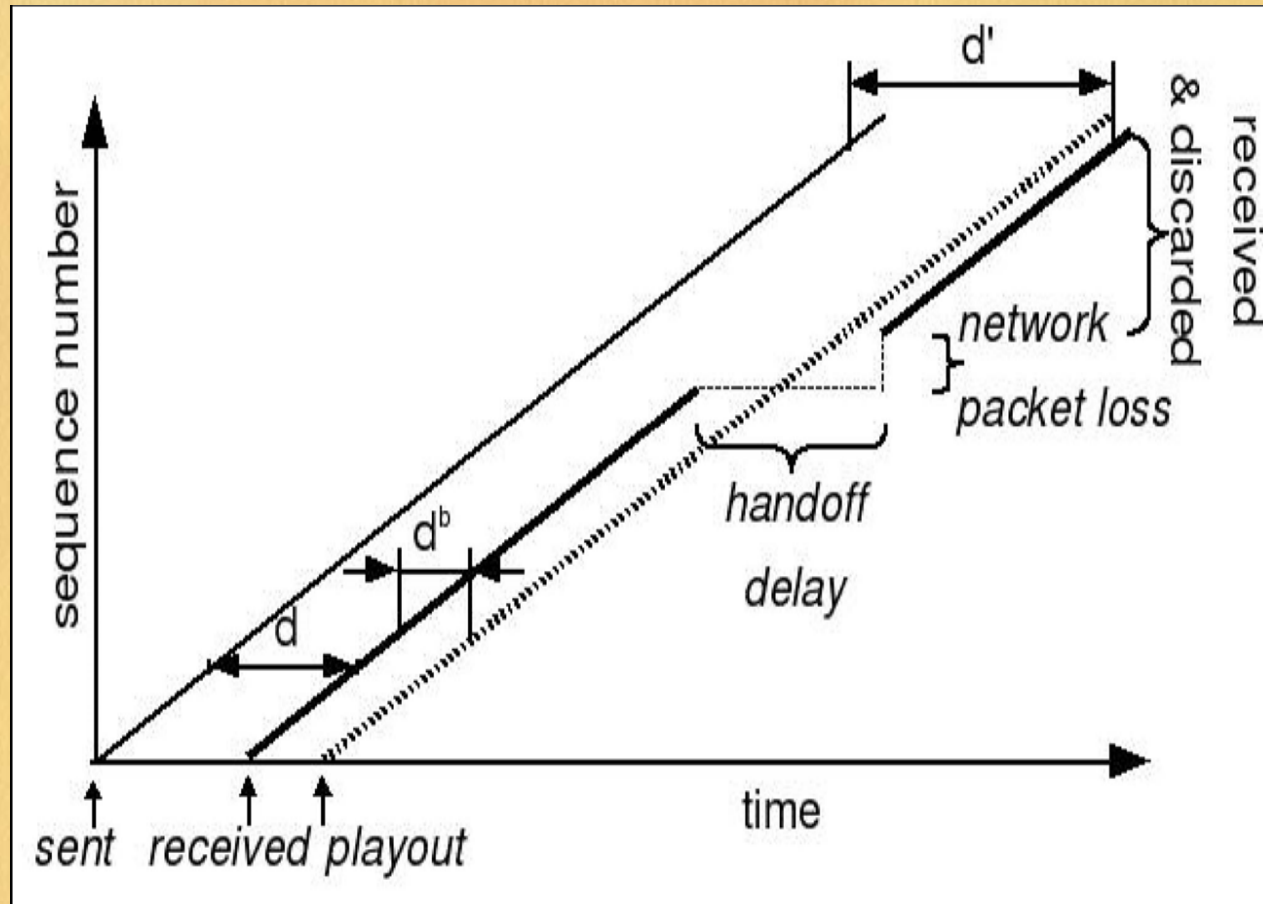


Fig: Packets discarding after a handoff towards a network with a larger trip delay

# ADAPTATION AT THE APPLICATION LAYER-

## Delay across handoffs

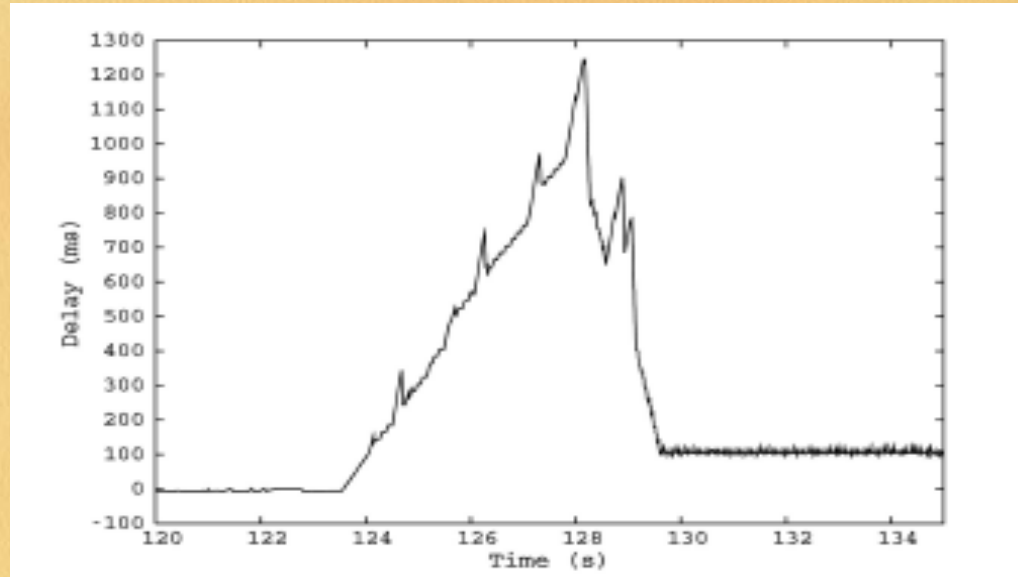


Fig: Delay during a handoff towards UMTS

- $d' - d$  is by measuring the delay of subsequent packets with respect to the first one
- UMTS network does not provide QoS support
- $t = 129.5$  s, the delay decreases and there is an accelerated delivery of late packets



# ADAPTATION AT THE APPLICATION LAYER-

## Delay across handoffs

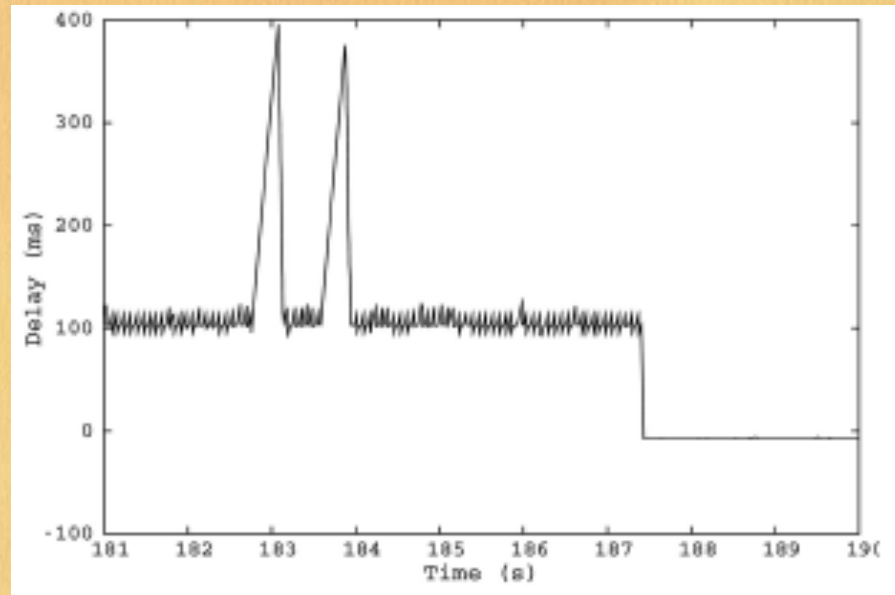


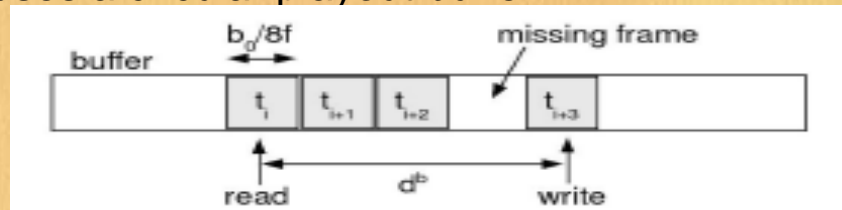
Fig: Delay during a handoff from UMTS

- Soft handoff(make-before-break) to WLAN (faster n/w) –small delay

# ADAPTATION AT THE APPLICATION LAYER-

## Buffer management for VoIP applications

- VoIP application uses a circular playout buffer B



- Network packet payload contains an audio frame of  $n$  audio samples
- Relation between playout time  $p$  and the location  $B[k]$  in the buffer
  - $p = 8k/b_o$
- To avoid jitter effect
  - Reproduce audio samples with a delay  $d_b$
- First packet written  $d_b b_o/8$ , play back time  $p_0 = d_b$
- $p^*$  - timestamp corresponding to the current read position.
  - Timely packets -written to the time  $p^* + d_b$ .
  - $p_i < p^*$  - discarded

$$p_i = p_{i-1} + t_i - t_{i-1}$$

# ADAPTATION AT THE APPLICATION LAYER

## Reactive algorithms

- Estimate network delay and its variance and use them to calculate the jitter buffer

$$\begin{aligned}\hat{d}_i &= (1 - \alpha)n_i + \alpha\hat{d}_{i-1}, \\ \hat{v}_i &= (1 - \alpha)|\hat{d}_i - n_i| + \alpha\hat{v}_{i-1},\end{aligned}$$

- Jitter Buffer

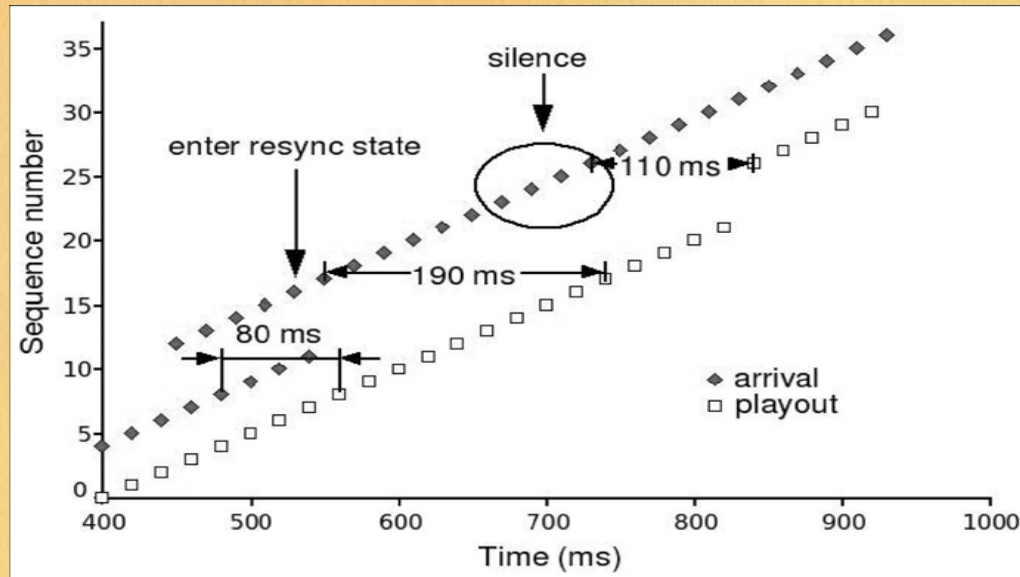
$$d_b = \beta\hat{v}_i$$

- $D_{\max}$  = upper limit for  $d_b$



# ADAPTATION AT THE APPLICATION LAYER

- ALGORITHMS MONITORING PACKET-LOSS RATIO AND BUFFER OCCUPANCY



# ADAPTATION AT THE APPLICATION LAYER

Fixed	$d_b$	avg loss (%)	avg delay (ms)	Fixed+MM	$d'_b$	avg loss (%)	avg delay (ms)
	80	49.00	77.12		80	49.12	22.27
	120	19.13	77.76		160	6.44	39.27
	160	6.32	106.01		240	4.17	80.11
	200	5.18	144.46		280	3.06	100.52
	240	4.05	182.48		320	2.07	121.07
	280	2.95	220.13		400	0.75	163.05
Monitoring	$d_b$	avg loss (%)	avg delay (ms)	Monitoring+MM	$d'_b$	avg loss (%)	avg delay (ms)
	20	5.93	32.87		20	5.91	33.26
	40	3.40	51.63		40	3.35	42.72
	80	3.20	91.27		80	2.91	63.03
	120	2.75	128.53		120	2.62	80.72
	160	2.08	163.58		160	2.21	100.48
	200	1.54	201.23		240	0.98	141.32
	240	1.00	235.87		280	0.54	159.23
Reactive	$\beta$	avg loss (%)	avg delay (ms)	Reactive+MM	$\beta$	avg loss (%)	avg delay (ms)
	1	8.47	31.30		1	6.78	31.93
	2	3.97	45.80		2	3.47	43.63
	3	2.41	60.30		3	1.93	55.00
	4	1.84	76.01		4	1.46	67.54

Figure: Performance of Playout Algorithms with and without MM

- 15 traces are collected at the MN, each containing 180 s of conversation 4 handoffs
- Monitoring -packet threshold is set equal to 3 and  $T_p = 60$  ms
- Reactive - $\alpha = 0.9$
- Fixed+MM- Handoff Notification to set  $d'_b = d' - d + \beta v$
- Monitoring +MM -  $d'_b = \beta v$

# ADAPTATION AT THE APPLICATION LAYER-

## Experimental results

- Fixed-size algorithms
  - MM adapts the size of the jitter buffer to the network delay
  - Noticeable reduction of buffering delay at the same loss rate.
  - Cannot adapt to spikes or delay variations that are not related to handoffs
- Monitoring algorithms
  - Adapt to variations of delay but not of jitter.
  - Fixed db results either too high in the WLAN phase or too low in the UMTS phase
  - Monitoring +MM algorithm overcomes this drawback by changing db
- Reactive algorithms +MM
  - Dynamically choose the value of  $\alpha$ .
  - value of set  $\alpha$  low when the jitter is high, and vice versa



# ADAPTATION AT THE APPLICATION LAYER-

## Experimental results

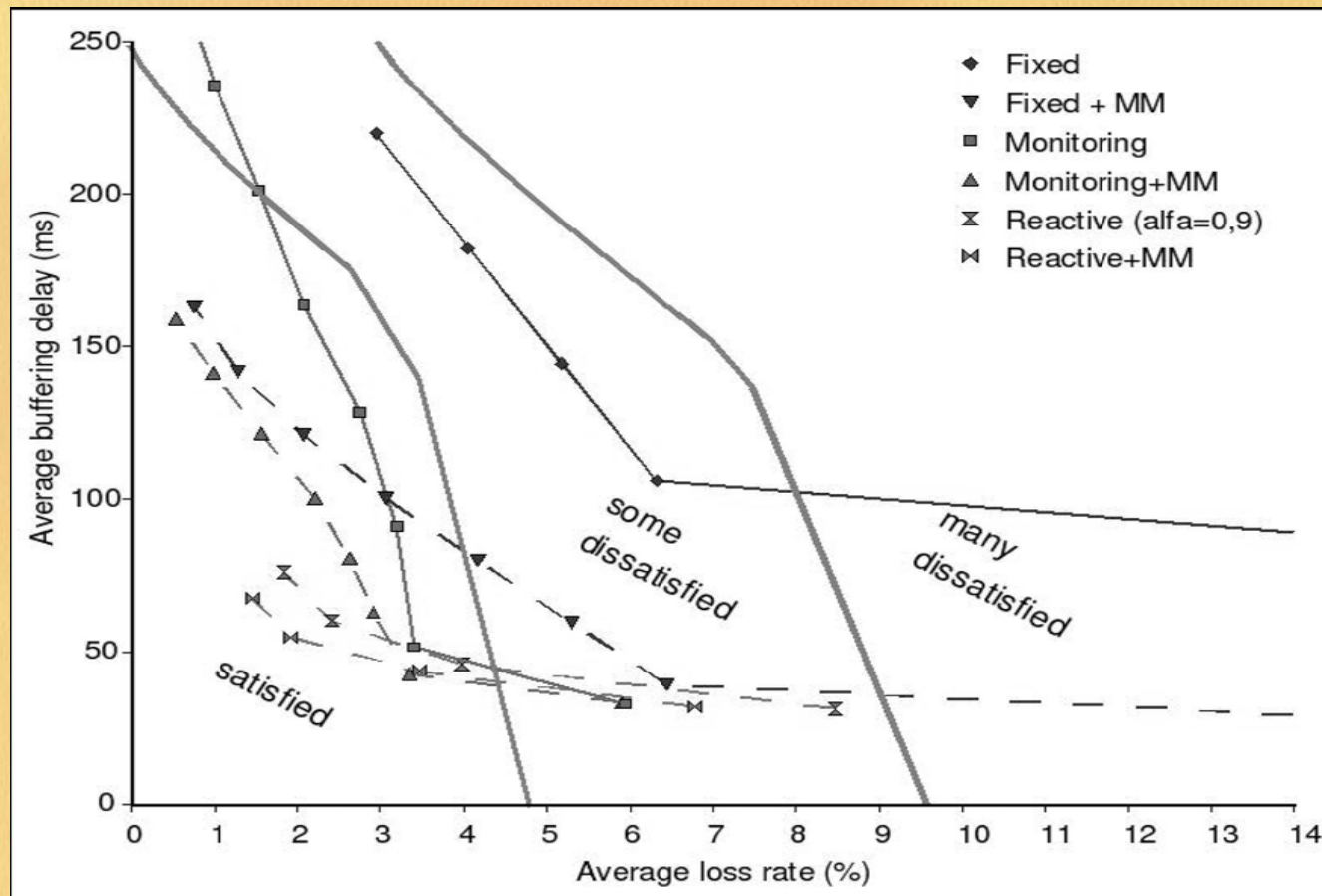


Figure: Packet-loss/delay ratio and user satisfaction(STSTQA) for adaptive algorithms

# ADAPTATION AT THE APPLICATION LAYER-

## Experimental results

	handoff WLAN-UMTS	handoff UMTS-WLAN
L3 handoff, adaptative playout algorithm, $d_b = 80$ ms	$1.85 \pm 0.34$	$1.67 \pm 0.62$
MM-assisted handoff, $d_b = 80$ ms	$2.29 \pm 0.33$	$2.38 \pm 0.38$
MM-assisted handoff and adaptation, $d_b = 40$ ms $d'_b = 120$ ms	$3.17 \pm 0.25$	$3.00 \pm 0.48$

Figure: Comparison of Mean Opinion Score Ratings (5 = Excellent, 1 = Bad)

- MOS is the Rating Scale
- Network losses are high
- Monitoring playout algorithm-Improvement on packet loss
- Monitoring algorithm that receives notifications from the MM - to change the jitter buffer

# CONCLUSION

- Cross-layer approach
  - Relies on Mobility Manager module on the mobile device to provide information about network parameters and handoff events
  - Effective for tuning parameters concerning audio stream playout and buffer management
- Adequate performance of a VoIP application in a mobility situation
  - Close to the one reported in literature on wired geographical networks
- No changes to existing protocol and network infrastructure